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Method and apparatus for attenuating external origin noise reaching the eardrum, and for improving intelligibility of electro-acoustic communications

The present invention relates to methods and apparatuses

5   for attenuating external origin noise reaching the eardrum, and for improving the intelligibility of electro-acoustic communications.

The technical field of the invention is the construction of acoustical protectors for the ear.

10 Passive soundproofing means are known such as headsets or earmuffs placed over the ears to protect persons required to remain in very noisy surroundings. Such means are used, for example, by workers working in certain types of workshop, by the drivers of very noisy vehicles (airplanes, tanks, ...), by

15 ground personnel at airports, etc...

Such headsets generally comprise a muff of absorbent material which envelops the ear.

In this type of sound insulation, incident airbourne sound waves are attenuated by reflection and by absorption in the

20 mass of the material which is acting as a passive screen.

Experience shows that passive soundproofing means are not very effective for low frequency sounds, and in particular in the frequency range below 500 Hz. In order to be effective at such frequencies, such headsets would require prohibitive

25 densities or thicknesses of material.

Sound insulating headsets are known which additionally include a loudspeaker or an electro-acoustic transducer incorporated inside each muff in order to allow the user to hear messages which are transmitted by the electro-acoustic

30 means.

The problem to be solved is to improve the efficiency of headsets and other passive soundproofing means by adding apparatus thereto for improving the attenuation of sounds of external origin lying in the frequency range where the passive

35 means is not very effective.

Apparatus known as an active acoustical attenuator is also known which serves to attenuate some sounds by making them

2

interfere with other sounds which are set up in phase
opposition to the sounds to be attenuated.

First attempts took place around 1953-1956 and have been
described by Olson and May.

5   The active acoustical attenuator proposed by these authors
comprises a microphone connected to a loudspeaker via an
electronic amplifier in such a manner that the loudspeaker
produces pressure within the cavity which opposes the pressure
due to the incident wave picked up by the microphone. If the
10 incident wave is random noise, the attenuation obtained by this
method is not very good, and in addition this method gives rise
to instabilities due to resonances at certain frequencies
("howlaround", or the Larsen effect).

In 1955-1956, Hawley and Simshauer published work inspired
15 by Olson's work. In order to avoid resonances due to sound
feedback, Hawley proposed making soundproofing headsets which
attenuated only some kinds of unwanted noise, which was either
in the form of pure harmonic sounds or else in the form of very
narrowband noise. Such headsets can only attenuate noise of a
20 form determined in advance and they do not attenuate noise
lying in a wide frequency range.

At pages 399 and 403 of Inter Noise 1983, Chaplin and
Smith describe an anti-noise device enabling harmonic sounds to
be attenuated by making them interfere with a synchronous sound
25 in phase opposition as generated by a synthesizer controlled by
digital electronics. Such apparatus using a sound which is
synchronous with the sounds to be attenuated can be used only
for attenuating a noise made up of a pure frequency together
with its harmonics. Any variations in the frequency of the
30 sound must be slow in order to make it possible for the digital
processing to correct the frequency of the feedback signal.
The need for a synchronization signal means that a noise
canceller must be provided for each source of noise. Digital
electronics is complex and expensive to implement. This method
35 is thus not usable for improving the efficiency of passive
soundproofing means intended to be used in very noisy locations
in sound fields essentially constituted by random noise.

French patent No. 75 34 024 (A.N.V.A.R.) describes active sound absorbing apparatus for attenuating plane sound waves propagating along a duct.

French patent 83 13 502 describes regulator apparatus for
5 an electro-acoustic system in which a filter is incorporated between the pickup member and the loudspeaker, said filter having a transfer function which is the inverse of the transfer function of the assembly constituted by the loudspeakers and the room in which listening is taking place.

10 The filter used is a non-recursive programmable digital filter of the convolution type provided with an input sampler.

This patent also described means for attenuating noise propagating along a guide, which means comprise a microphone connected to a plurality of loudspeakers by an electronic
15 system including a filter of the convolution digital filter type which causes the electrical signals to be appropriately filtered in order to minimize the noise.

The means described in these prior documents require major adaptation in order to be usable in combination with portable
20 passive soundproofing means such as headsets or earplugs.

The latter prior document describes active acoustical attenuators each including a digital filter which provides a complex transfer function which is the inverse of the transfer function of an electro-acoustic system. Such a solution
25 including a digital filter requires a processor unit to be used in real time and constitutes an expensive solution which is difficult to use with walkabout passive soundproofing means.

U.S. Patent document US-A-4,494,074 (Bose) describes earphones including a microphone and an electro-acoustic transducer
30 connected via a feedback loop including a preamplifier, a signal adder, compensation circuits including filters, and an amplifier for driving the transducer.

British Patent document GB-A-2 160 072 (Plessey) describes active noise attenuators associated with earphones. Each attenuator
35 comprises a microphone connected to an electro-acoustic transducer via a feedback loop including an inverting amplifier.

4

German Patent document No. DE-A-2 925 134 (Sennheiser Electronic K.G.) describes active apparatus for providing protection against external noise, the apparatus comprising a microphone connected to an electro-acoustic emitter via a feedback loop

5 shown diagrammatically, including an amplifier and optionally including an adjustable high-pass or low-pass filter.

Some of these prior documents mention the presence of a filter in the feedback loop, but none of them specifies how said filter is chosen and what transfer function it should have

10 in order to obtain the best possible attenuation.

An aim of the present invention is to provide active attenuation means associated with means delimiting a cavity around the ear, said attenuation means including a filter whose transfer function is determined on the basis of the open loop

15 transfer function of said cavity in such a manner as to obtain good active attenuation over a wide frequency range, and in particular including the low frequency sounds for which passive soundproofing devices are not very effective.

The method in accordance with the invention for

20 attenuating external origin noise reaching the eardrum is of the type in which each ear is associated with passive soundproofing means which together delimit a cavity, with a microphone and an electro-acoustic transducer being disposed inside said cavity, said microphone and transducer being

25 interconnected by a feedback loop including a constant gain amplifier and a filter to constitute an active soundproofer.

The aim of the invention is achieved by means of a method in which the transfer function $C(w)$ of the filter is a complex polynomial function and the open loop transfer function $H(w)$ of

30 the assembly constituted by the transducer, the microphone, and the cavity delimited by said passive soundproofing means and the ear is measured, the coefficients of said polynomial function $C(w)$ then being calculated so that the product of the constant gain of said amplifier multiplied by the modulus of

35 said open loop transfer function $|H(w)|$ and by the modulus of the transfer function of said filter $|C(w)|$ is much greater than unity over the range of frequencies where said passive

soundproofing means is of low effectiveness while retaining stability in the feedback system.

Apparatus in accordance with the invention for attenuating external origin sounds comprises passive soundproofing means for delimiting a cavity around each ear and additionally comprises a microphone and an electro-acoustic transducer which are disposed inside said cavity and which are interconnected by a feedback loop including a constant gain amplifier and a filter with which they constitute an active sound attenuator.

The aim of the invention is achieved by means of apparatus in which the transfer function $C(w)$ of the filter is a complex polynomial function such as the product of the modulus $|C(w)|$ multiplied by the modulus $|H(w)|$ of the open loop transfer function and by the gain K of the amplifier is greater than unity throughout the range of the frequencies to be attenuated.

The invention provides means enabling the efficiency of headsets and other similar passive soundproofing means to be improved by associating them with an active sound attenuator including, in its feedback loop, a filter for optimizing the efficiency of the passive soundproofing means over a wide frequency range.

It is known that passive soundproofing means are poor attenuators of low frequency sounds, i.e. sounds at frequencies below 500 Hz, which sounds are very common in some noise, for example vehicle engine noise.

Apparatus in accordance with the invention improves sound attenuation, _inter alia_, by means of a feedback loop including a constant gain amplifier and a filter which is preferably an analog filter of the polynomial type whose components are designed so that the ratio between the total acoustic pressure inside the cavity and thus at the eardrum to the acoustic pressure due to external noise that has passed through the passive soundproofing means remains low over the entire range of frequencies that are to be attenuated.

The design of analog polynomial filters is a technique that is well known to the person skilled in the art.

The modulus of the overall transfer function of a feedback loop in accordance with the invention is equal to the product of the constant gain (K) of the amplifier multiplied by the modulus of the filter transfer function $|C(w)|$ and by the modulus of the open loop transfer function $|H(w)|$, i.e. as measured between the input to the transducer and the output from the microphone.

The modulus and the argument of the open loop transfer function H(w) can be measured and plotted by injecting a white 10 noise signal into the transducer input and by simultaneously applying said signal and the signal emitted by the microphone to a spectrum analyzer.

Given the modulus and the argument of the open loop transfer function of a passive attenuation means equipped with 15 an electro-acoustic transducer and with a microphone having a given disposition, it is possible to calculate the values of passive components for a filter such that the modulus of the transfer function of the filter follows a given law over the frequency range to be attenuated while still satisfying the 20 stability criterion so as to avoid producing any resonance due to "howlaround" or the Larsen effect.

Apparatus in accordance with the invention serves equally well for attenuating continuous noise and impulse noise, i.e. noise of the kind generated by shocks or detonations, giving an 25 amplitude that varies very fast. This is made possible by the fact that the electronic processing takes place in real time.

The following description refers to the accompanying drawings which show non-limiting embodiments of the invention.

Figure 1 is a diagrammatic overall view of soundproofing 30 apparatus in accordance with the invention.

Figure 2 is a section through an ear fitted with apparatus in accordance with the invention.

Figure 3 is a diagram of the components of apparatus in accordance with the invention.

35 Figure 4 is a graph showing the modulus of the sound spectrum inside a given passive headset when the external noise is white noise.

Figure 5 is a graph showing the attenuation obtained using a device in accordance with the invention.

Figure 6 is a diagram showing the components of a device in accordance with the invention for enabling messages to be
5   heard as transmitted over an electro-acoustic path.

Figure 7 is a section through an ear fitted with apparatus in accordance with the invention.

Figures 8 and 9 show specific embodiments of a filter.

Figure 1 is a diagram showing apparatus in accordance with the invention placed over the ears of a subject for the purpose
10 of attenuating external noise as perceived by the subject.

This apparatus comprises passive soundproofing means constituted, for example, by two muffs 1d and 1g which surround each ear and which are interconnected by a headband 2 in order to constitute a headset.   The muffs 1d and 1g are applied
15 against the sides of the head and together therewith they delimit respective cavities each enclosing a corresponding ear pinna.   Each such cavity is closed and comprises two cavities: the first corresponding to the inside of the muff; and the second corresponding to the auditive cavity delimited by the
20 pinna, the external ear duct, and the eardrum.

The muffs 1d and 1g constitute passive soundproofing apparatus for reflecting and absorbing a portion of the soundwaves, thereby attenuating the noise which reaches the ear.   The muffs 1d and 1g may be replaced by any other passive
25 soundproofing apparatus, for example by plugs placed at the entrance to each external ear duct.   In this case, the closed cavity is constituted by a first half-open half-cavity comprising the earplugs and its miniaturized transducers, together with a second half-cavity which is constituted by the
30 eardrum and the external ear duct.

Experience shows that passive soundproofing apparatus attenuates low frequency sound poorly.

Figure 4 is a graph with sound frequency plotted along the X axis and with the modulus of the transfer function plotted
35 along the Y axis, for the case of a microphone placed inside a passive muff as shown in Figure 1 and external white noise.

It is recalled that the transfer function H(w) is a complex function expressing the ratio between the Fourier transform S(w) of a signal leaving a device and the Fourier transform E(w) of the signal applied to the input to said device, i.e. H(w) = S(w)/E(w).

Figure 4 thus shows the noise level in relative decibels as measured inside the cavity delimited by one of the muffs 1d or 1g at each frequency contained in the external noise. It can be seen that at frequencies below 600 Hz, the attenuation obtained is less good than at higher frequencies.

The problem to be solved is to add active soundproofing means to passive soundproofing apparatus in order to provide attenuation mainly at those frequencies which escape from the passive attenuation, given that the active means may also improve attenuation at higher frequencies that are already attenuated by the passive means.

Figure 2 is a section through an ear showing the external ear duct 3, the eardrum 16 and the pinna 4 placed inside an insulating muff 1 including a lining 5 of cellular material which is pressed against the skin around the pinna.

Figure 2 shows an electro-acoustic transducer 6, e.g. a small loudspeaker, which is supported by a partition 7 fixed on the insulating muff 1. The emitting face of the loudspeaker 6 is directed towards the ear. It is placed substantially opposite the opening to the external ear duct 3 and preferably at a short distance from said opening, e.g. at a distance of about a few centimeters. The transducer 6 is connected by an electrical conductor to a terminal A.

Figure 2 also shows a microphone 8 which is disposed inside the external ear duct 3 or between the emitting face of the loudspeaker 6 and the inlet to said duct 3, and which is connected via a conductor to a terminal B.

Figure 1 shows a housing 9 which contains the electronic components and the circuits respectively connecting terminal A to terminal B for one of the ears and terminal A' to B' for the other ear.

Figure 3 is a theoretical diagram showing a section through the closed cavity 10 corresponding to coupling the open cavity of the muff with the open cavity of the ear, and in which an electro-acoustic transducer 6 and a microphone 8 are

5 placed. The microphone 8 is connected to the transducer 6 via an electronic circuit comprising an amplifier 11 having constant gain K and an active type of analog filter 12, i.e. a filter comprising selective amplifiers associated with passive components (capacitors or resistors).

10 Figure 3 shows the terminals A and B visible in Figures 1 and 2.

Interference takes place in the external ear duct between two opposing acoustic pressures.

The first pressure is due to noise coming from the outside

15 through the muff 1 and attenuated to a greater or lesser extent depending on its frequency. After applying the Fourier transform, this acoustic pressure may be represented by a complex function Po(w) where $\underline{w}$ is the angular frequency corresponding to each frequency $\underline{f}$ (i.e. $w = 2\pi f$).

20 The second pressure is a pressure resulting from the waves emitted by the transducer 6 on the basis of signals emitted by the microphone 8, as amplified by the amplifier 11 and transformed by the filter 12. The circuit running from the microphone via the amplifier, the filter, and the loudspeaker, and

25 returning to the input of the microphone constitutes a feedback loop which is closed within the external ear.

Let P(w) designate the complex function representing the Fourier transform of the resulting pressure.

Let K designate the constant gain of the amplifier.

30 Let H(w) designate the open loop transfer function between points A and B.

Let C(w) designate the transfer function of the filter 12.

Given that the feedback loop is closed, the total pressure P(w) may be written, using the Fourier transforms, as being

35 equal to the sum of the incident pressure Po(w) and the pressure due to the feedback which is equal to: K.H(w).C(w).P(w).

This gives rise to the following equations:

$$(1) \quad P(w) = P_0(w) + K.H(w).C(w).P(w)$$

whence

$$(2) \quad P(w)/P_0(w) = 1/(1-K.H(w).C(w))$$

5 This equation shows that the acoustic pressure P(w) reaching the ear may be attenuated over a given frequency range if the ratio P/Po can be reduced throughout this range, i.e. if it is possible to obtain a value for the complex product K.H(w).C(w) which is much greater than unity, while nevertheless avoiding

10 resonance phenomena at certain frequencies.

There would be no point in attenuating noise over a range of frequencies if this were also to give rise to even more inconvenient interfering noise due to the howlaround effect.

An examination of equation (2) shows that if it is possible

15 to provide a filter having a transfer function $C(w) = H^{-1}(w)$ over the entire given frequency range, then all that would be required would be an amplifier having a gain K which is very high in order to obtain very good noise attenuation over the frequency range.

20 It is possible to approximate to a transfer function which is the inverse of the open loop transfer function by using digital filters associated with a processor unit, however this solution is bulky, expensive, and is not capable of working in real time as is required by the feedback system.

25 The filter 12 used in the apparatus shown in Figures 1 to 3 is an analog filter which is small and cheap, and which is not capable of providing a transfer function which is the inverse of the open loop transfer function.

The open loop transfer function H(w) may be measured by

30 omitting the housing 9 and applying an electrical input signal to terminal A corresponding to white noise and by picking up from B the output electrical signal emitted by the microphone 8.

These two input and output electrical signals should be then simultaneously applied to a spectrum analyzer programmed

35 to perform analog-to-digital conversion on both signals and to compute the open loop transfer function H(w) corresponding to discrete frequencies.

The spectrum analyzer includes a screen on which it displays, as a function of frequency, both the phase of the transfer function and the variations in its modulus.

Spectrum analysis shows that the open loop transfer
5   function, i.e. the ratio between the Fourier transform of the output signal at point B and the input signal at point A depends largely on the shape and the volume of the cavity 10 and also on the respective positions of the transducer 6 and the microphone 8 relative to the cavity.

10   Laboratory studies have shown that the phase shifts and the variations in the modulus of the open loop transfer function can be reduced.

Instead of using the open loop transfer function H(w) directly, this transfer function may be optimized prior to
15   electronic processing. Optimizing a transfer function which exists in a feedback loop makes it possible to obtain greater active acoustic attenuation over a wider frequency range. Optimizing the open loop transfer function comes to the same as performing "pseudo-linearization" thereof, such that the
20   modulus and the phase of H(w) are constant and the phase shift is small over the range of frequencies to be attenuated in accordance with the principle of the invention. Thus the greater the extent to which the transfer function H(w) is "linear" the more the electronic processing for the feedback is
25   simplified.

The method proposed for optimizing the open loop transfer function H(w) consists in a series of steps. First, the cavity 10 is subdivided into two cavities by means of a partition 7 (cf Figures 2 and 3, for a loudspeaker having little or no
30   baffling) such that the "front cavity" corresponds to the assembly constituted by items 3, 4, 5, 6, 7, and 16, while the "rear cavity" corresponds to the assembly formed by items 1, 6, and 7. The microphone 8 is then placed in the above-described "front cavity" either in front of the loudspeaker and close to
35   the inlet to the external ear duct, or else inside the duct. The microphone is then placed at a short distance from the loudspeaker in order to reduce phase shift, and the volume of

the cavity 10 is reduced as much as possible in order to avoid resonance and anti-resonance effects.

In practice, since the open loop transfer function H(w) may be quite different depending on the geometrical shape of the enclosure 10 and on the positions of the transducer 6 and of the microphone 8, any improvement to the soundproofing provided by a passive muff 1 of given shape and nature by means of an active system begins by determining and fixing the positions of the transducer 6 and the microphone 8 inside the muff 1, and then in using a spectrum analyzer to measure the open loop transfer function H(w) of this set of items when placed on an ear.

Figure 7 is a diagrammatic section through another embodiment of apparatus in accordance with the invention. Parts which are equivalent to parts of Figure 2 have been given the same references.

In the Figure 7 embodiment, an annular part 15 delimiting an intermediate cavity 15a is interposed between the partition 7 and the ear pinna 4 inside the lining 5. Advantageously, this part 15 serves as a support for the microphone 8 which may be disposed in a hollow in the part 15 as shown in Figure 7, or which may be juxtaposed with said part 15.

Depending on the ranges of frequencies to be attenuated, the open loop transfer function H(w) may be optimized by means of acoustic filtering. For example, if low frequencies are to be attenuated, an annular part 15 fixed against the partition 7 provides acoustic filtering by a cavity effect. In the Figure 7 example, the ratios of the diameters, the thicknesses, the aperture of the screen (baffle) 7 of the annular part 15, and of the external ear duct 3 define a low-pass acoustic filter. It is thus possible to model the transfer function H(w) by calculating and performing acoustic prefiltering prior to electronic processing.

The dimensions of the intermediate part 15 and of the intermediate cavity 15a delimited thereby are determined so that the ratio between the dimensions of said intermediate cavity and of said front and rear cavities separated by the

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partition 7 give rise to acoustic filtering having a pass band corresponding to the range of frequencies to be attenuated.

Preferably, the active face of the microphone faces towards the emissive face of the loudspeaker in order to
5 optimize the open loop transfer function H(w).

Once the transfer function H(w) has been established, the filter 12 is designed to provide a transfer function C(w) such that the product of the constant gain K of the amplifier 11 multiplied by the modulus of the open loop transfer function
10 H(w) and by the modulus of the transfer function C(w) of the filter 12 is much greater than unity within the range of frequencies over which an improvement in external noise attenuation is desired. This condition is not sufficient. It is also necessary to verify that the stability criterion is
15 satisfied in order to avoid "howlaround" phenomena which would lead to the production of noise by virtue of resonance at certain frequencies (the Larsen effect).

In order to avoid howlaround due to resonance, it is necessary for the Nyquist stability criterion to be satisfied
20 as is well known to the person skilled in electrical art. It is briefly recalled that the Nyquist criterion consists in verifying on a so-called Nyquist plot that the overall transfer function F(w) of all of the items in a system does not intersect the real axis at a point greater than unity for all
25 of the frequencies in the spectrum being listened to.

The Nyquist plot has the real portion of the transfer function plotted along the X axis and the imaginary portion plotted along the Y axis.

Let F(w) be the overall open loop transfer function of the
30 system.

Let $|F(w)|$ be the modulus and $\phi(w)$ be the phase of said function.

Let $\Delta\phi$ be the phase stability margin and $\Delta\rho$ be the modulus stability margin.

35 The phase stability margin $\Delta\phi$ corresponds to the amount, in radians, by which the phase of the transfer function F(w) may vary due to unforeseeable interference delays without the system becoming unstable.

The modulus stability margin $\Delta\rho$ corresponds to the amount of unforeseen variation in the modulus of the transfer function F(w) which can be tolerated in the feedback system without it becoming unstable.

It has been shown that active acoustic attenuation is obtained over a range of frequencies wi by means of an electro-acoustic feedback loop without giving rise to an unstable system because of resonance if, and only if:

$$|F(wi)| > 1 - \Delta\rho \quad \text{and}$$

$$2K\pi + \Delta\phi < (wi) \ 2K\pi - \Delta\phi \quad \text{for } K = 0,1,2,\ldots$$

If $|F(wi)| \leq 1 - \Delta\rho$

then the system provides no active attenuation, in which case the system is stable regardless of the value of the phase $\phi(w)$

Providing both conditions are satisfied, those acoustic waves which are not stopped by the passive soundproofing apparatus 1 and which reach the inlet to the external ear interfere with acoustic waves emitted by the loudspeaker 6, and this interference minimizes the ratio P/Po between the modulus P of the resulting acoustic wave and the modulus Po of the incident acoustic wave over the entire range of frequencies for which the product $K.|H(w)|.|C(w)|$ is much greater than unity. In addition there are no interfering sounds due to howlaround.

In practice, the filter 12 used is preferably an active analog filter comprising, for example, one or more integrated circuit filters each having a polynomial transfer function of the form:

$$C(w) = (a1(w)^2 + a2(w) + a3)/(b1(w)^2 + b2(w) + b3).$$

Such a filter includes resistors and/or capacitors capable of being connected to the terminals of the integrated circuit and whose values are adjustable in order to obtain predetermined values for the real constant coefficients a1, a2, a3, b1, b2, and b3 of the function C(w).

The calculations necessary for constructing polynomial filters and the forms of the transfer functions of such filters are well known to persons skilled in the electrical art.

In general, the coefficients of the denominator b1, b2, and b3 are fixed in advance and they determine the cutoff

frequency and the Q factor of the filter, while the coefficients a1, a2, and a3 of the numerator are varied in order to determine the nature of the filter.

If a1 and a2 are equal to zero and a3 is different from zero, a low pass filter is obtained which is advantageous since it provides a transfer function having a large modulus at low frequencies. However, such a filter gives rise to phase rotation which passes simultaneously through 0° and through ±180° at a frequency which is less than the cutoff frequency where the modulus $|C(w)|$ is > 1, thereby giving rise to howlaround which happens in the present case at 0° if K is positive.

If a2 and a3 are zero and a1 is not zero, a high pass filter is obtained which is not advantageous since the sounds which most require attenuating are low or medium frequency sounds.

If a1 and a3 are zero and a2 is not zero, a pass band filter is obtained which introduces phase rotation, e.g. from +90° to ∓90° without going through 0°, and through ±180° when the modulus of the filter, i.e. $|C(w)|$ is at its maximum, which is advantageous since there is no danger in the present system of that giving rise to howlaround.

In practice, when it is mainly low frequency sounds that are to be attenuated, it is preferable to use a mixture of filter types combining the effects of a bandpass filter and of a low-pass filter. It is also possible to associate a plurality of mixed type polynomial filter circuits (combining high-pass, low-pass, and bandpass types) in parallel or to associate bandpass filters only.

Figure 5 is a graph having audible frequencies plotted in logarithmic co-ordinates along the X axis and having acoustic pressure levels plotted in decibels along the Y axis. The solid line curve Po represents the spectrum of acoustic pressure Po measured when using only a passive soundproofing headset which attenuates frequencies less than 1500 Hz poorly. The dot-dashed curve P represents the acoustic pressure spectrum measured when using the same headset in association

with an active attenuator in accordance with the invention.   In
this case the electronic circuit used is a single active analog
filter of the bandpass type.

The acoustic pressures were measured in the presence of
5   white noise emitted by a loudspeaker enclosure over the
frequency range 20 Hz to 20,000 Hz.   The acoustic pressures are
measured by means of a small microphone which is inserted into
the external ear as close as possible to the eardrum and they
thus represent the sounds actually perceived by the ear.   The
10   spectra Po and P are obtained by means of a spectrum analyzer
which applies a Fourier transform to the measurements.

The Figure 5 graph shows average results measured over
very many tests.   This graph shows that attenuation lying
between 0 dB and 8 dB is obtained in the frequency band 20 Hz
15   to 55 Hz, with very good attenuation lying between 0 dB and 50
dB being obtained in the frequency band from 65 Hz to 2000 Hz,
and with slight attenuation being obtained between 3800 Hz and
20,000 Hz.

There are two narrow frequency bands between 55 Hz and
20   65 Hz and between 2000 Hz and 3800 Hz in which there is no
attenuation and in which there is even a slight increase in
noise level, which increase remains below 6 dB.

Advantageously, a plurality of bandpass filters are
connected in parallel in the feedback loop, for example second
25   or third order filters having juxtaposed pass bands which cover
the major portion of the frequency range in which open loop
phase rotation of the system including the filters satisfies
Nyquist stability criterion.

This parallel connection of a plurality of filters has the
30   advantage of enabling the individual pass bands of the filters
to be added together while giving rise to phase rotation which
is the average of the phase rotations of the individual filters
and which thus remains between -90° and -270° for second order
filters and between -45° and -315° for third order filters,
35   i.e. within regions satisfying the Nyquist stability criterion.

In a preferred example, a bandpass filter is used in
conjunction with a high-pass filter having the same cutoff
frequency and the same Q factor.

This result is obtained by using an integrated circuit in association with variable resistors which can be adjusted to set the three coefficients a1, a2, and a3 of the numerator of the polynomial transfer function.

5   It is thus possible to obtain either a predominantly bandpass filter by giving more weight to the coefficient a2 relative to the other two coefficients a1 and a3, or else to obtain a predominently low pass filter by giving more weight to the coefficient a3.

10   The effect of the high-pass filter contained in such a compound filter is to reduce phase rotation and to make it tend towards zero at high frequencies without substantially altering the appearance of the modulus, thereby enabling the width of the attenuation passband to be enlarged towards high

15   frequencies without introducing instability and thus increasing the attenuation and the width of the attenuated band.

Instead of associating three filters having the same cutoff frequencies and the same Q factors, it is also possible to connect a plurality of low-pass, bandpass, and high-pass

20   filters having different cutoff frequencies such that the lowest cutoff frequency is that of the high-pass filter and the highest cutoff frequency is that of the low-pass filter, and with the filters having different Q factors.

This association gives rise to sharper edged pass bands

25   than the previous association with phase rotation tending towards zero outside the pass band, it also gives rise to higher frequencies in the low frequency range.

All of these adjustments can be performed by initially performing digital simulation using the measured open loop

30   transfer function H(w) and by manually adjusting the variable resistors as needed in order to determine the optimum transfer function C(w) of the filter.

All of these examples clearly show the wide range of possibilities offered by the method in accordance with the

35   invention for obtaining high performance active sound attenuators which may be implanted in existing soundproofing headsets or in headsets which have been specially designed to receive such attenuators.

Curve P1 of Figure 5 shows an example of the attenuation curve obtained by associating a plurality of second order analog filters of the low-pass, bandpass, and high-pass types in parallel with different cutoff frequencies and with

5   different Q factors. It can be seen that attenuation is obtained over a frequency band which is wider than that shown by curve P and that the peaks for which noise level is increased are eliminated.

Very often, sound insulating headsets include an

10   incorporated loudspeaker in order to transmit messages to the user by electro-acoustic means.

If a headset in accordance with the invention including an active attenuator is fitted with a second loudspeaker which is not included in the feedback loop and which is intended for

15   transmitting messages, sounds emitted by the second loudspeaker are considered as sounds coming from the outside and are therefore picked up by the microphone and by the feedback loop and are attenuated. This solution must therefore be avoided.

Another solution consists in applying the electrical

20   signals conveying the message directly to the loudspeaker 6 which forms a part of the active attenuator.

Equations and practice show that such a circuit enables a message to be heard even though the message is distorted by the feedback. However, there is a better circuit.

25   Figure 6 is a diagram similar to Figure 3 and shows this circuit. Portions thereof equivalent to portions of Figure 3 are given the same references.

The housing 9, in addition to the constant gain amplifier 11 and the filter 12 also contains a summing circuit 13 which

30   receives firstly the signals from the microphone 8 and secondly electric signals from an external transducer 14 for conveying a message.

Experience shows that such a circuit makes it possible not only to attenuate the noise reaching the eardrum without

35   attenuating the level of the message, but also to escape from the transfer function of the loudspeaker 6 and of the cavity 10 over the range of frequencies under consideration.

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In the following equations, K, C(w), Po(w), Pt(w) and H(w) have the same meanings as in equations (1) and (2).

The open loop transfer function H(w) between points A and B may be decomposed into a product of a first transfer function

5   H1(w) between point A and the input to the microphone multiplied by a second transfer function H2(w) which is the transfer function of the microphone. In practice, the modulus of the microphone transfer function may be substantially constant over a wide frequency band providing the microphone is

10 of good quality.

Let Sp(w) designate the Fourier transform of the electric signal emitted by the external transducer 14 for conveying the message.

The following equation may then be written:

15 (3) $Pt(w) = (K.H1(w).C(w).Sp(w)+Po(w))/(1-K.H(w).C(w))$
     If the gain K of the amplifier is very large, then:

    (4) $Pt(w) \simeq Sp(w).(H1(w)/H(w)) + \varepsilon \simeq Sp(w).H2(w) + \varepsilon$
     Equation (4) shows that the Figure 6 circuit serves both to attenuate the residue of external origin noise by virtue of

20 the feedback loop while simultaneously improving hearing of the message which is no longer subjected to the transfer function H1(w) which represents the transfer function of the combination of the cavity and the loudspeaker.

The Figure 6 circuit can be used to provide a kind of

25 feedback loop filtering of a message which is mixed with external noise.

If the response of the microphone 8 is considered as being ideal, i.e. $H1(w) \simeq H(w)$, then equation (4) becomes:
    $Pt(w) \simeq Sp(w) + \varepsilon.$

30 The suggested circuit does not reduce the level of the message Sp(w) which is thus not only reinstated intact, but in addition, it is freed from any transfer function, and this would not be the case if the signal leaving the transducer 14 were applied directly to the loudspeaker 6.

35 It can thus be seen that the solution in accordance with Figure 6 makes it possible to hear a message sent electro‐ acoustically and transmitted via the same transducer 6 used by

the active attenuator after increasing its level relative to
the level of outside noise and also improving the message by
virtue of the fact that the transfer function H1(w) which
inevitably appears in any other circuit is eliminated.

5   Equations and experimental measurements show that any
other position for the summing circuit in the feedback loop
does not improve the intelligibility of the message by "feed-
back filtering" and leads to degradation of the intelligibility
of the transmitted message, which degradation can sometimes be
10   very high. The intelligibility of the message can be further
improved in accordance with the Figure 6 circuit providing the
range of frequencies to be attenuated is centered on the speech
band 300 Hz to 3000 Hz. This range of frequencies corresponds
to the most sensitive region of the human ear, and thus when
15   active attenuation is being used above all for improving
message intelligibility, it is no longer necessary to provide
high attenuation at low frequencies as is the case when
providing protection only without communication. Indeed, it
becomes necessary to improve passive attenuation from 300 Hz by
20   broadening as much as possible the range of attenuated
frequencies in order to reach 2000 Hz or better still 3000 Hz,
even if some efficiency is lost in pure noise attenuation.

The above description relates above all to low and medium
frequency ranges. In this case, equations show that the
25   combination of acoustic attenuation and of "feedback filtering"
of the transmitted message improves working conditions and also
message intelligiblity. However, it has just been shown that
if the frequency range of active attenuation is centered on the
speech band, i.e. 300 Hz to 3000 Hz, even if little attenuation
30   is obtained in the high frequency zone thereof, the intelligibil-
ity of the transmitted message is better still and this effect is
further improved if the range of attenuated frequencies extends
up to high frequencies of the order of 2000 Hz to 3000 Hz.

Equation (3) also shows that if Po(w) is very small, the
35   feedback filtering phenomenon of the message is retained since
the second member of the equation is independent of the
pressure Po(w) and the result of equation (4) for gain K large

in comparison with unity is thus again satisfied, i.e. the message Sp(w) is free from any transfer function. The active acoustic absorption system can thus be used solely for improving the message by feedback loop filtering, even when

5 external noise is not inconvenient.

The above description relates to preferred embodiments in which the filter 12 is an analog filter. However, it is specified that it would also be possible to use a digital filter on its own or in association with a processor unit, in

10 combination with passive soundproofing apparatus when used on board a vehicle.

In a variant embodiment, an active apparatus in accordance with the invention includes a small microphone which is placed in the external ear together with a miniaturized transducer

15 whose rear face includes a coating which constitutes a plug and which is engaged in the inlet to the external ear. In this case, the cavity is reduced to the volume delimited by the external ear, the eardrum and the transducer-carrying plug, and the open loop transfer function H(w) is highly linear so that

20 it is easy to obtain a good level of attenuation over a wide frequency band by electronic filtering.

CM We claim:

claim 1

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